The NEVATON BPT - "Blumlein-Pfanzagl-Triple" 3-capsule Stereo- and Surround-Microphone with Center-Zoom Function: ready for 5.1, 7.1 and beyond …

USER MANUAL Vers. 1.3

Ser. # ……

0. Preamble – "Why build a 3 capsule microphone ?"

As the name already indicates, the "Blumlein-Pfanzagl-Triple" (BPT) makes use of a third figure-of-eight capsule, which is added to the well known Blumlein-Pair of two figure-of-8 capsules, which have an included angle of 90° and are oriented at +/- 45° towards the sound source. The new capsule is pointed directly towards the middle of the sound source at 0°.

Why add a third capsule at all, as the Blumlein-Pair already captures 360° ? The Blumlein-Pair arrangement is appreciated by many as 'very naturally sounding' (see [Streicher and Dooley, 1985]), which is attributed also to the fact that the signals of the two channels are largely decorrelated (meaning: very different) also at low frequencies, which is important for good spatial impression. This is a unique feature which is almost exclusively reserved to crossed figure-of-eights (when it comes to 'one-point' or closely spaced microphone techniques), as all other microphone patterns (with hypercardioids as an exception) do not provide similar signal-separation for low frequencies.

As a one-point recording technique with highly directional microphones the BPT-mic is characterized by providing precise localization.

Tomlinson Holman described this behaviour, which can also be found with figure-of-eight microphones, with the following words: „ … The system [i.e. Blumlein-pair] aims the microphones to the left and right of center; for practical microphones, the frequency response at 45° off the axis may not be as flat as on axis, so centered sound may not be as well recorded as sound on the axis of each of the microphones …" (see [Holman, 2000]).

This is of course one of the main reasons why it makes very much (sonic) sense to add a third figure-of-eight microphone or capsule to the "Blumlein-Pair" arrangement. In addition, this gives the sound engineer a high degree of freedom, of how broad / (i.e. 'stereophonic') or narrow ('monophonic') he would like the sound image to be, by simply varying the level of the Center-capsule in relation to the L and R capsule.
For normal recording applications, the level of the center microphone will usually be set between -6 dB to -3dB in relation to the L and R microphone.
Deviations of this are – of course – possible and for some applications it might make sense to set it at -10dB or – if it is necessary to emphasize the center of the sound-source – it may be set to 0dB or even more.

In order to shield off unwanted signals from the rear it is preferable to place an acoustic barrier (for example in the form of a sound-absorptive acoustic panel) behind the BPT microphone. This helps to achieve a better direct-/diffuse sound (i.e. direct signal /reverb) ratio and enables the sound engineer to move the system further away from the sound-source thereby usually achieving also a better balance between instruments (or instrumental groups) in case of larger ensembles.
(Rem.: If the absorptive panel cannot be used for optical reasons, the three capsules of the microphone can be switched to cardioid pattern and attenuate sound from the rear in this manner. However – if possible, the use of an absorptive panel is recommended, as this will result in a much clearer sonic image achieved in the recording, since diffuse sound arriving from the back is being eliminated more effectively)

1. Using the Center-Capsule C to "zoom in" on the sound source

Rising the level of C 'focuses' or 'zooms in' to the centre of the sound source, while – as a side effect, desirable or non-desirable – increasing the 'monophonic' part of the overall signal. Theoretical calculations (using the "Microphone Assistant V.2" software - see [Wittek, 2002]) have shown that if C is set at the same level as the L and R microphone the result will be a gradual increase and boost of volume by 3 dB towards the middle of the sound-source.
This can be of advantage in connection with broadcast-applications (news, location recording, etc.), as it is common practice to work with (highly) directional microphones in order to create an "acoustic focus".
If the center capsule's signal is adjusted to a relative level of -10dB in relation to the L and R capsule, almost no emphasis or gain is added to the sounds coming from the centre of the sound-source.

It is of advantage to use the additional fig-8 center-capsule, as it helps to achieve a more stable sound-image especially if sound sources with complex (i.e. frequency dependent) radiation characteristics (as is the case with many music instruments) need to be recorded.

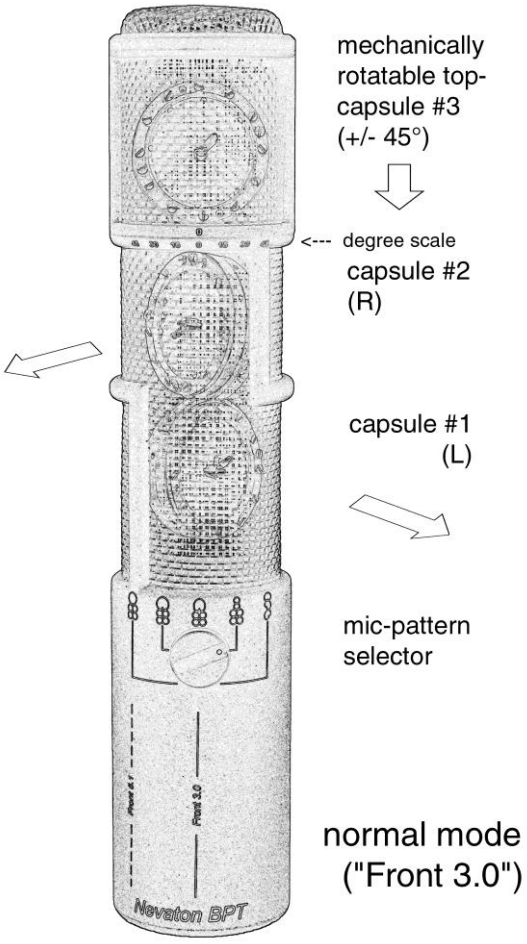


Fig. 1: BPT-mic (normal mode)

2. Use of a HPF on the Center-Capsule (for enhanced de-correlation)

Results of various researchers have shown that low signal-correlation (cross correlation) is necessary at frequencies below approx. 200Hz in a stereo (or binaural) signal in order to have good spatial impression as a listener (see –among others - [Hidaka et al, 1995] and [Griesinger, 1998]).
Due to the 90° physical offset angle between the L and R capsule, which have fig-8 characteristics, the signal of these two will be completely de-correlated over the entire frequency response (to be precise: for diffuse sound signals, arriving from all directions). Adding the signal of the center capsule (by mixing it in on the stereo-bus, or routing it to the center-speaker in a 5.1 surround setup) signal perception will certainly become more correlated over the entire frequency band. For the sake of good spatial impression, correlation for bass-frequencies should be kept at a minimum: the simplest way to achieve this is by setting a High-Pass-Filter (HPF) on the signal of the Center capsule, with a cut-off frequency somewhere in-between 90 and 160Hz, depending on the frequency response of the sound-source, the acoustics of the room, etc.

This can be done on the mixing desk, or already while recording by use of the High-Pass Filter-control (fc=90Hz) on the rear of the BPT-mic set to the position on the right.

3. Choosing the right distance between BPT and sound-source

Concerning the normal "Blumlein-pair" microphone technique Tomlinson Holman stated that "… The system makes no distinction between front and back of the microphone set, and thus may have to be placed *closer* than other coincident types … ". There is certainly a point to it, as in a normal concert situation the rear lobes of the fig-8 microphones will mainly pick up reverb from the hall and therefore the direct/diffuse sound ratio will be affected in a negative way.
If the natural mix of these two signal components is too reverberant, an absorptive panel should be placed behind the BPT.
When using an absorptive panel, the soundfield of the rear hemisphere is kept away from the BPT and therefore the critical distance (i.e. reverberation radius; see appendix) is effectively doubled for the microphone, which means it can be moved away from the sound-source by the amount equivalent to the reverberation radius, while the direct/diffuse sound ratio will remain the same (due to the addition of the absorptive panel behind the BPT).

A second important factor needs to be considered for the placement of the BPT microphone, which is the physical opening angle of the L and R fig-8 capsules (angled at +/-45° relative to the sound source). Even though the Blumlein-Pair is said to have a correct recording angle of 120° (see [Streicher and Dooley, 1985]), it is certainly preferable if the main body of the sound source stays well within the included angle of 90°, as otherwise there is the danger that a sound source in front on the far left might be picked up by the rear lobe of the right fig-8 capsule of the Blumlein Pair and therefore will be replayed not only from the left side (left capsule signal), but also from the right side, but with a phase inversion.

In [Faulkner, 1981] the question of optimal microphone positioning in order to achieve a good balance between all instrumental groups of an orchestra in a concert location has been discussed. As sound engineer Tony Faulkner points out the "old rule of thumb" to position the main (stereo) microphone at a distance of about half the width of the orchestra away from the sound-source can be enlarged to the whole width in case of highly directional microphones, which is the case of fig-8 microphones (and even more so, if an absorptive panel is used). Therefore a microphone position in the 6^{th} up to the 9^{th} row, as pictured in fig.2, might not only be "good seat" position for a listener, but also for a Blumlein-Pair or BPT-mic.…

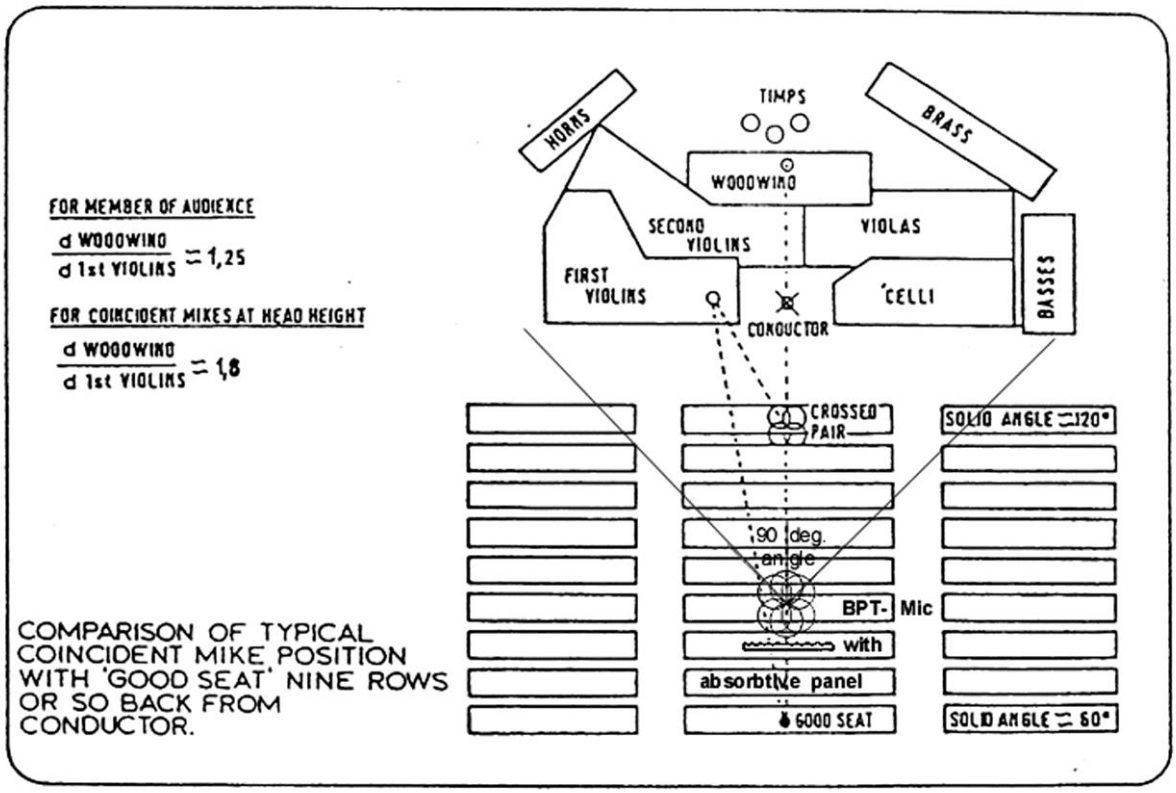


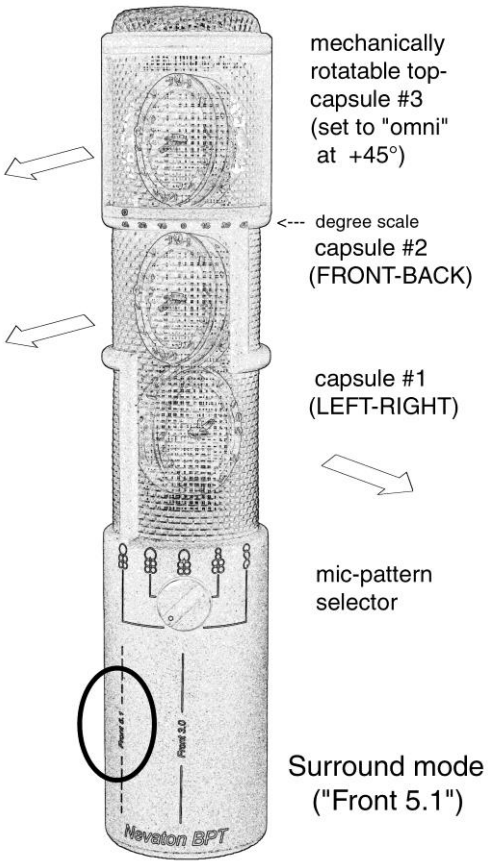
Fig. 2: Microphone position vs. listener position in a concert hall [adapted from Faulkner, 1981]

4. RECORDING IN SURROUND Version 1: The BPT in 5.1 mode (see **Fig. 3** ->)

As stated by audio engineer Tomlinson Holman "… Two channel stereo can produce the sensation of looking into a space beyond the loudspeakers; multichannel stereo can produce the sensation of being there ..." [Holman, 2000]

This is a strong statement for making surround recordings and the BPT makes this quite easy, as no external processor is needed:

If you want to create a 5.1 surround signal with just one BPT-mic, the pattern selector needs to be switched to total left position. In this mode the top capsule (# 3) is in omni mode (and rotated to +45°), while capsules #1 and #2 remain in fig-8 mode. As the 5.1 signal is going to be derived by using MS-decoding, the BPT-mic as a whole needs to be rotated by 45° so that the lowest capsule (#1) becomes a figure-8, which picks up sound to the LEFT and RIGHT of the microphone, while capsule #2 picks up the FRONT and BACK sound-field.



To help visually with the right orientation of the mic-body, there is dashed line indicating the correct 0° alignment of the microphone: ----FRONT 5.1 -----

As can be seen in the scheme of figure 4, the 5.1 surround signal is then created by combining (mixing, i.e. summing) the signals of the omni-capsule (#3) with the fig-8 capsules (#1 and 2#) in an appropriate way.

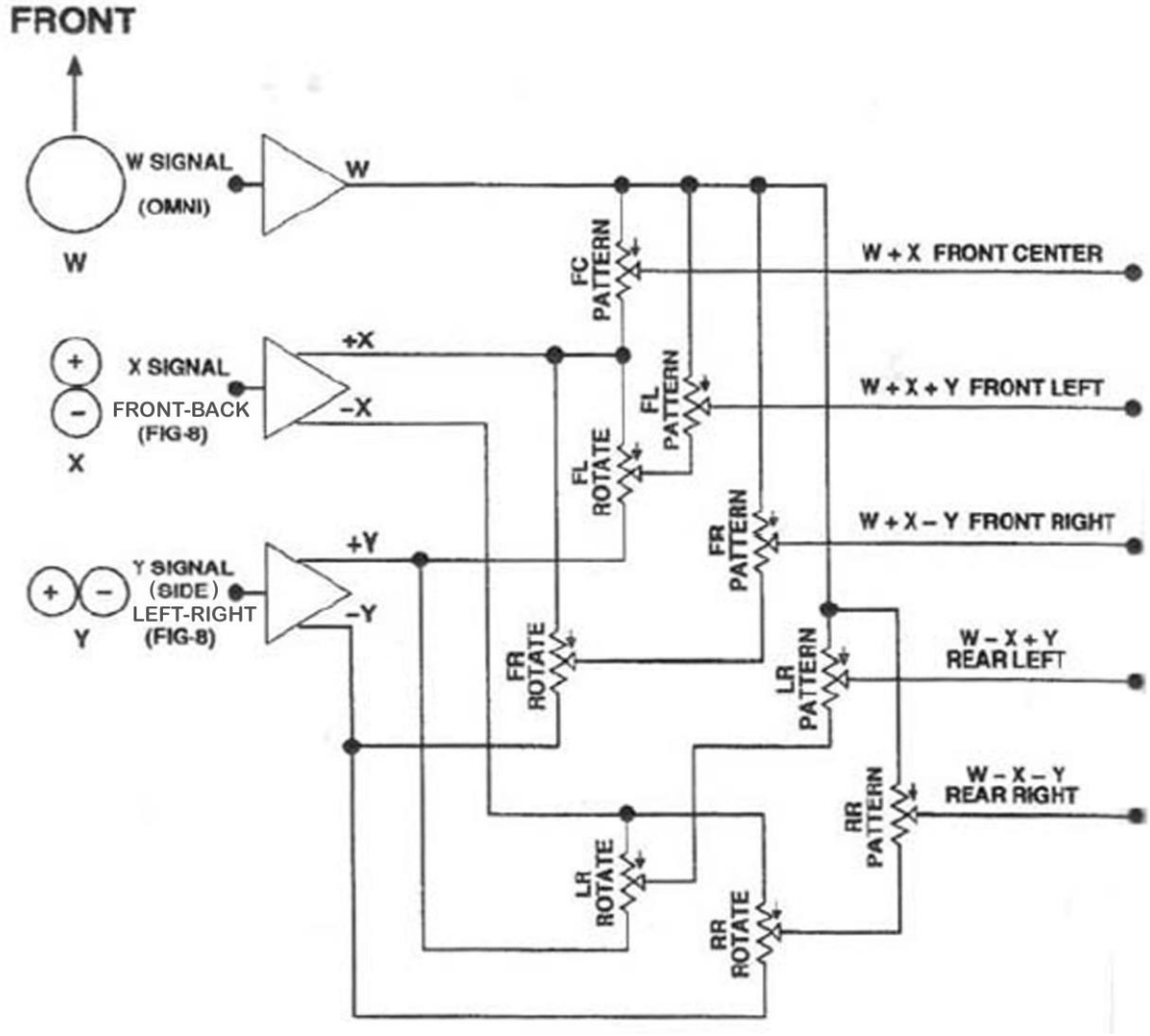


Fig. 4: signal processing schematic for 3-capsule microphone (adapted from [Eargle, 2004])

For the 5.1 “Center-channel” the signal of the omni-capsule (#3, “W”) and front-back capsule (#2, “X”) need to be combined: W + X
For “Front Left” all the signals W + X +Y need to be summed.
For “Front Right” the same three signals, but on Y (capsule #1) the phase needs to be reversed by 180°: W + X - Y
For “Rear Left” (also called ‘Left Surround’ or LS) we need to perform: W – X + Y
For “Rear Right” (also called ‘Right Surround’ or RS): W – X - Y
The processing as seen in fig. 4 – which might seem complicated at first – can very easily be achieved on a normal mixing desk: on digital mixing desks, which allow to assign the signal-input of a channel A/D-converter to multiple channel strips, this is particularly easy.

The graphic below shows a routing assignment on an 8-bus console which mixes the W, X and Y signals on the appropriate busses for 5.1 recording or reproduction.

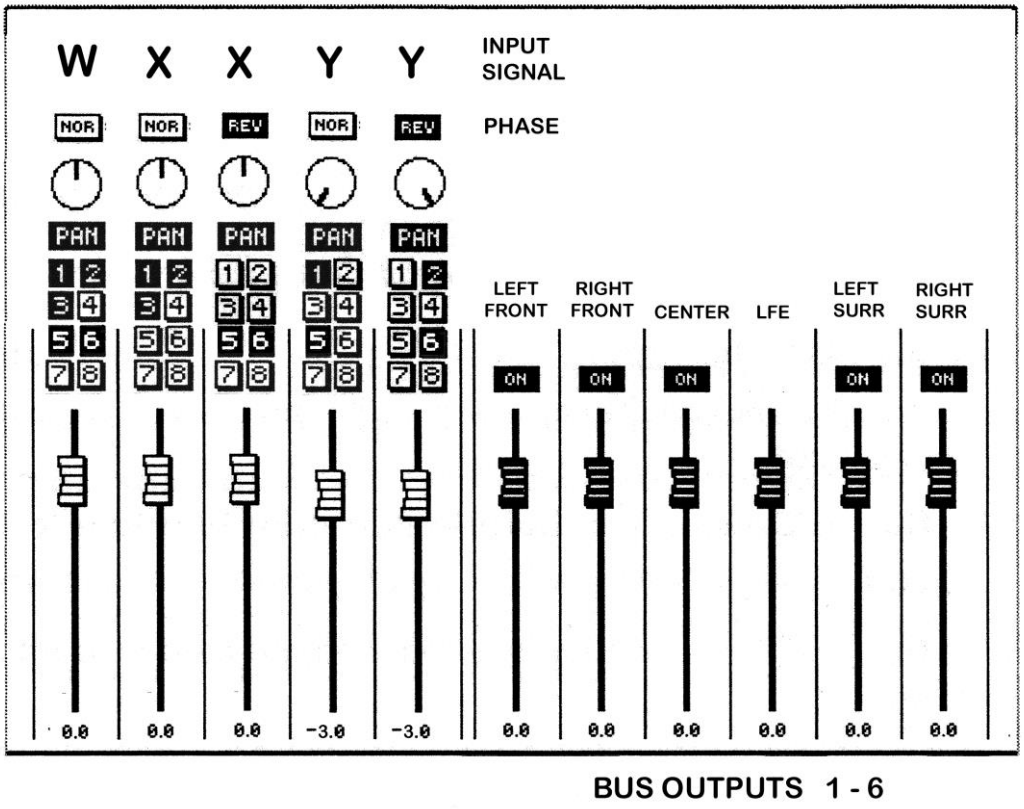


Fig. 5: BPT-microphone signal MS-decoding on an 8-bus mixing desk

The routing scheme above also makes use of the PAN function on a mixing desk, which is not really needed if the desk allows “direct routing” to single busses. In that case all input faders can be set to 0dB and the PAN-control at center.
The -3dB level attenuation on the fader needs to be done if for L/R-routing on stereo-busses the PAN-control needs to be employed, as a signal which is panned in this way has a 3dB gain in respect to signals which are routed to both bus-channels with the PAN-control set at center (due to the ‘panning law’ which is normally employed in mixing desks).

Creating an LFE-signal:
The signal-routing layout shown in fig. 5 does not provide an LFE-signal; should this be needed, the easiest way to generate one would be to bring up the W (omni) signal on a free input-channel, then low-pass filter it with a cut-off frequency set somewhere between 90 - 120Hz, route it to bus #4 and balance it in level wise, according to taste.

5. RECORDING IN SURROUND - Version 2: two BPT’s in “Back-to-back” mode (2 x 3.0)

There is a second possibility to achieve a 5.1 surround recording with the BPT microphone, which offers better channel separation (especially between front and rear signals) by use of 2 BPT microphones and (at least one) acoustic baffle and absorptive panel in-between.
In this case for both front and rear a BPT-mic in 3.0 mode is being used, which directly deliver the signals for L, Center, Right (Front) and LS, RS (Rear).
This version has the advantage that no signal-processing (MS-decoding) is needed.

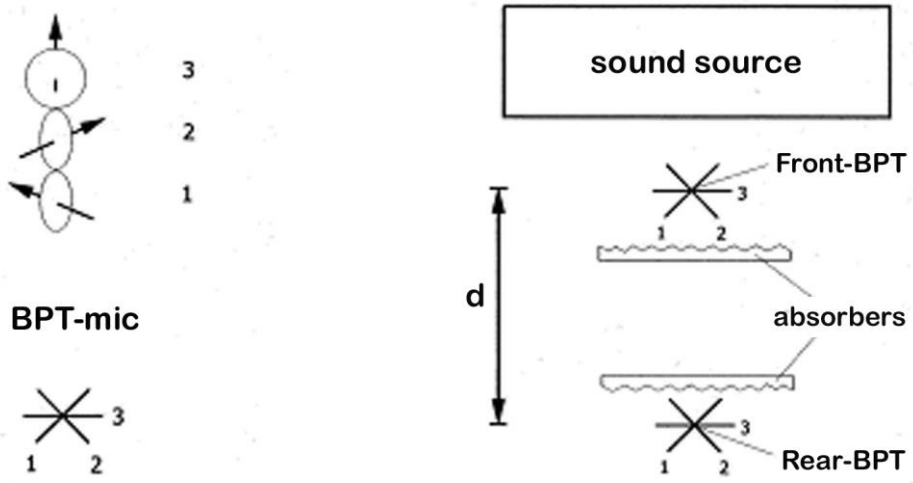


Fig. 6: Two BPT microphones in 'Back-to-back' configuration with absorber panels (for surround rec.)

The distance "**d**" between the two BPT microphones can be varied to control the amount of spatial impression contained in the recording: if **d** is in the range of 50cm to 100cm spatial impression will be less, if **d** is in the order of several meters the front and rear signals will be more de-correlated and spatial impression will be larger. However, a distance of 10m should not be exceeded between the front and rear BPT in order to keep time-of-arrival differences for sound below 30ms, as above this limit echo effects might occur, depending on the room acoustics and placement of the BPT- microphones in relation to the sound source.

6. Using the Figure-of-Eight pattern to "re-balance" the orchestra (and soloists)

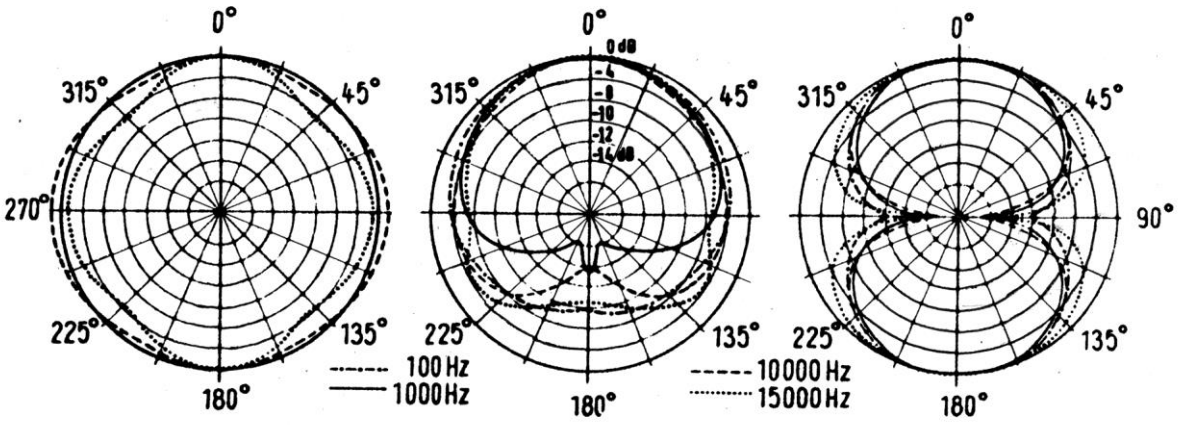


Fig. 7: Polar diagrams of a double membrane condenser mic (Neumann KM88)

As can be seen in fig. 7 off-axis level attenuation is much more pronounced with fig-8 microphone patterns, than with cardioids: while a cardioid will have only 1-2dB attenuation at 30° and approx. 4 dB at 60°, the fig-8 pattern already exhibits a 9dB level drop at 60°,

which - due to its polar pattern – increases dramatically above 60° to reach 'infinity' (total attenuation) at 90°.
This strong directional characteristic of the fig-8 pattern can be used as an advantage if one needs to re-balance the instrumental groups (including soloists) of an orchestra:

For this purpose it is recommended to point the main axis of the fig-8 towards the sound element which needs the most support level-wise. In the example of Figure #8 below, the mic is directed towards the woodwinds, but due to the much closer distance to the first desks of the sting section, these will still have an advantage of 2dB over the woodwinds.

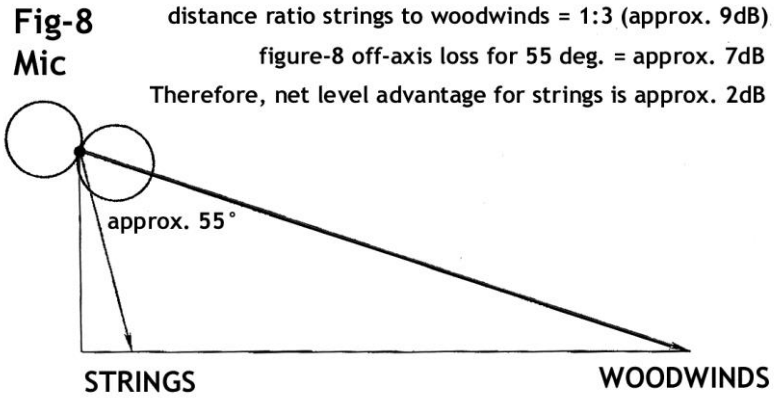


Fig-8
Mic
distance ratio strings to woodwinds = 1:3 (approx. 9dB)
figure-8 off-axis loss for 55 deg. = approx. 7dB
Therefore, net level advantage for strings is approx. 2dB
approx. 55°
STRINGS
WOODWINDS

Fig. 8: vertical attenuation (angle of incidence) as a creative tool for level balancing

In a typical situation, pictured in the diagram above, the microphone could be positioned approx. 3m (9feet) above the first desks of the strings, while the woodwinds might be approx. 9m (27 feet) away. Hence a distance ratio of 1:3 applies. As can also be seen in the graphic concerning the "reverberation radius" or "critical distance" in the appendix, direct sound suffers a level drop of 6dB, when the distance is doubled (2:1). Doubling the distance again (4:1) would lead to another level drop of 6dB, so a total of 12dB attenuation; a distance ratio of 1:3 therefore results in a level drop of 9dB.
[Rem.: usually, when miking the strings from above they might tend to sound rather bright and sometimes aggressive, therefore a slight level-drop (and also high-frequency attenuation) due to the off-axis position of the strings might even be an advantage. However, we might be able to achieve a more balanced overall sound by moving the BPT main-mic further out in the hall]

If we imagine a situation of a hall with a reverberation radius D_{crit} = 6m for example, we might feel encouraged to move the main microphone (BPT) further back in the hall (with or without acoustic panel behind), so it might be slightly less than 6m away from the first strings and therefore approx. 12m away from the woodwinds.
If we decide to tilt the main axis of the fig-8 down a bit so that it is directed towards the middle between the woodwinds and the first desks of the strings, we would get the same amount of off-axis level attenuation for both of them, while the level drop due to distance would result in an advantage of 6dB for the strings in comparison to the woodwinds.
The further we move the BPT away from a large sound source like an orchestra, the more rounded or 'naturally balanced' the sonic picture becomes usually and the more effectively we can use the right orientation of the main-axis to feature sound elements within the

sound source, who need that kind of support. In the absence of spot-microphones, soloists in front of the orchestra next to the conductor, or a harp further back in the orchestra might be the usual candidates which need such help ...

To give an example of the effectiveness of diffuse sound attenuation due to the use of an acoustic panel behind the BPT microphone: in the concert hall of the Salzburg Festival (D_crit=5,5m) – in order not to obstruct sight-lines from the balcony - the BPT mic with panel is 12m (!) away from the conductor's podium. Since the curved absorptive panel shades off the rear half-hemisphere of diffuse sound, the critical distance is effectively doubled, so even with such a large distance to the orchestra, the direct-/diffuse-sound ratio is still a "healthy" one and provides nice-sounding results for high-quality documentary recordings ...

7. Flat absorptive panel vs. curved panel

As already mentioned above, in many recording situations the final result will benefit from the use of an absorptive acoustic panel behind the BPT-microphone in order to re-balance the direct- to diffuse-sound ratio (Rem.: … for use of the BPT in '3.0 mode'. If the BPT is used in the '5.1 mode', of course the rear side should not be covered or shielded off potential sound sources or diffuse sound).

A curved acoustic panel will usually be more effective, as with the same surface it manages to 'protect' a larger angle of the microphone from picking up unwanted sound. On the other hand, some focusing effects may be noticed with a curved (rounded) acoustic panel, if the absorptive materials are not fully absorptive over the whole frequency range (which is hard to achieve sometimes if its mechanically supportive elements are made of metal).

In practical experiments it has been found that curved absorptive panels, with diameters useful in connection with the BPT microphone (diameter approx. 40cm), may produce unwanted reflections (and therefore sound coloration due to comb-filtering effects) in the frequency band roughly between 500 – 700 Hz. In this case a band-filter tuned to about 600Hz with Q=3 to 5 and an attenuation of -4 to -8dB (inserted on at least the L and R BPT-mic signal) will help to eliminate unwanted sound coloration artifacts.

Fig.9: BPT with rounded panel

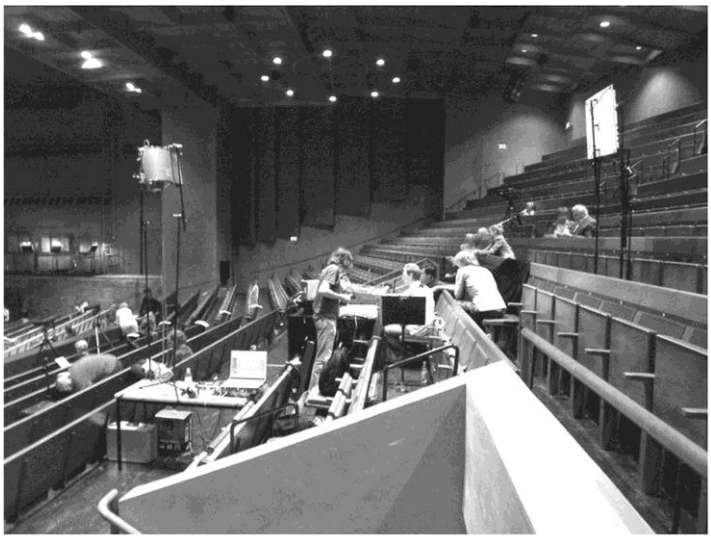


Fig. 10: BPT surround-recording with 2 BPT-mics in 'back-to-back' mode in auditorium (approx. 5m spacing) with rounded panel (front) and flat acoustic panel (back)

APPENDIX:

Definition of "critical distance" D_{crit} :

D_{crit} [m] = 0.057 sqrt (V [m^3] / RT60 [s])

or

D_{crit} [ft] = 0.03147 sqrt (V [ft^3] / RT60 [s])

with 1ft = 0.3048m or 1m = 3.2808 feet

D_{crit} …. "Critical distance" (i.e. reverberation radius)

sqrt …. square root

V …. Volume of the room

RT60 … reverb time of the room in seconds

The "critical distance" (or reverberation radius) from the sound source is reached, when diffuse sound reaches the same level (volume) as the direct sound.

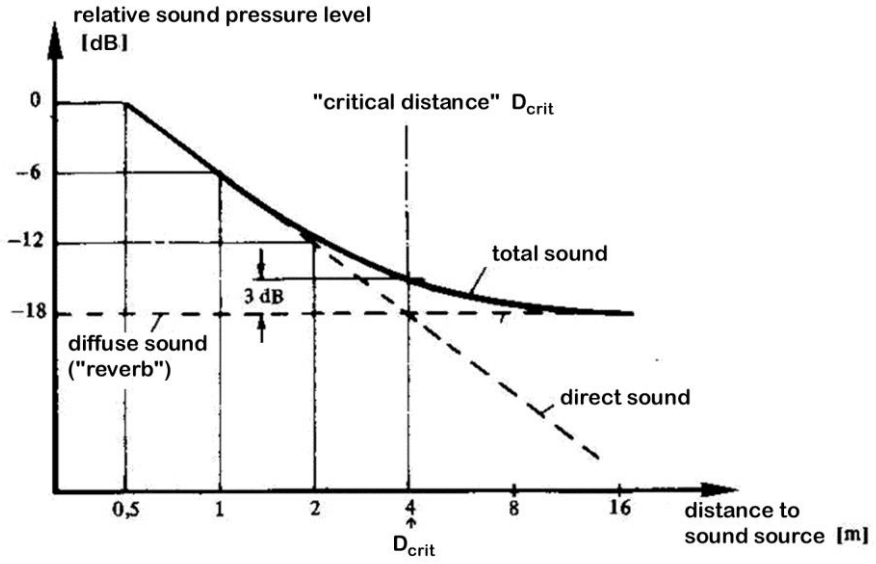


Fig. 11: "Critical Distance" (D_crit): Relation between SPL and distance to sound source for direct- and diffuse-sound in a room

Hall	**Volume** [m3]	Volume [ft3]	**Seats**	**RevTime** [s]	**D_crit** [m]	D_crit [ft]
Ottobeuren (church, Germany)	130000	4590300	n.a.	7,00	7,77	25,48
Royal Albert Hall, London	86650	3059612	5080	2,40	10,83	35,53
Amsterdam Concertgebouw	18700	660297	2206	2,00	5,51	18,08
Boston Symphony Hall	18740	661709	2631	1,80	5,82	19,08
Vienna Musikverein	14600	515526	1680	2,05	4,81	15,78
Teatro della Scala (opera, Italy)	11252	397308	2289	1,24	5,43	17,81
Eisenstadt Castle - concert-hall	6800	240108	n.a.	1,70	3,60	11,83
King's Theatre, London	4550	160661	n.a.	1,55	3,09	10,13
Brahmssaal, Vienna Musikverein	3390	119701	604	1,63	2,60	8,53
Esterháza Castle - concert hall	1530	54024	n.a.	1,20	2,04	6,68
control room (typ. rec. studio)	118	4167	n.a.	0,24	1,26	4,15

Table 1: Reverb time and resulting critical distance for performance spaces of different size

High Pass Filter Control on rear side of BPT-Microphone:

On the rear side of the BPT a filter-selector can be found:
A High-Pass Filter with corner frequency at 90Hz can be switched in for all three capsules, if needed.
Switch in left position, all three capsules linear.
Switch in center position: HPF for all 3 capsules
Switch in right position: only top capsule #3 has HPF activated

Literature references:

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Hidaka, T., Beranek L., and Okano, T. (1995): "Interaural cross-correlation, lateral fraction, and low- and high-frequency sound levels as measures of acoustical quality in concert halls", *J. Acoust. Soc. Am*. 98 (2), August 1995

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NEVATON BPT-Mic - technical data:

- Frequency Range: 20-20000Hz
- Free Field Sensitivity at 1000Hz and 1000 Ω: 20 +/-2 mV / Pa (or better)
- Equivalent SPL DIN/IEC 651: 5 dB-
- Max. Sound Pressure Level [for THD 0,5%]: 142dB
- Dynamic range: 137 dB
- Nominal output impedance: < 50 Ω
- Recommended Load Impedance: >= 1000 Ω
- Capsule diameter: 33 mm
- Microphone body diameter: 48 mm
- Length: 245 mm
- Weight: 808 g
- Phantom power (48V +/- 4V)
- Power consumption: approx. 10mA per capsule
- Connector Type: XLR, 7-pin (male connector built into microphone)

Pin assignment: 1= GND, pin 6+7 = signal „Left Capsule (#1)“ +/-, pin 4+5 = signal „Right capsule (#2)“ +/-, pin 2+3 = signal „Center capsule (#3)“ +/-